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Minimizing Packet Loss Using Buffer Management Scheme for Video Streaming

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ABSTRACT: Media streaming is a technique for transferring data so that it can be processed as a steady and continuous stream. Before the transmission across the network, the data frames are divided into packets. Due to the application level encoding schemes, there is an interpacket dependency in the streaming applications. In order to create efficient communication only if all the packets are received at the receiving side. In the streaming applications if it is drop a single packet there is useless in the delivery of whole sequence. So, in order to overcome this problem, we use Selective frame discard analyze their performance by means of competitive analysis. The QoS of a video stream is measured in terms of a cost function, which takes into account the discarded frames. But in this method due to congestion or heterogeneous nature of the network, data loss may be occurring. So, in order to overcome this problem, we introduce an innovative technique called buffer management algorithms in specific environments. In such environments, the algorithm used to decide which packet to drop in case of buffer overflows must be carefully designed, to avoid goodput degradation. We present a model that captures such inter packet dependencies, and design algorithms for performing packet discard. Traffic consists of an aggregation of multiple streams, each of which consists of a sequence of interdependent packets. So, by using this buffer management algorithm we effectively reduce the packet loss and maximize the Quality-of-Service.

KEYWORDS: Buffer Management, Interpacket dependency

I. INTRODUCTION

Wireless network [1] refers to any type of computer network that utilizes some form of wireless network connection. It is a method by which homes, telecommunications networks and enterprise installations avoid the costly process of introducing cables into a building, or as a connection between various equipment locations. Wireless telecommunications networks are generally implemented and administered using radio communication. This implementation takes place at the physical level of the OSI model network structure.

Fig. 1 Wireless icon

The emergence of high-speed networks facilitates many multimedia applications that rely on the efficient transfer of compressed video. Such applications include streaming video broadcasts, distance learning, shopping services, etc.

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However, compressed video, especially variable-bit-rate (VBR) [2] video, typically exhibits significant burstiness on multiple time scales, owing to the encoding schemes and the content variation between and within video scenes. This burstiness complicates the design of efficient transport mechanisms for such media. In a network where resources such as bandwidth and buffering capacity are constrained there is a need for an efficient video delivery system that can achieve high resource utilization and maximize the QoS perceived by the user.

A simple strategy called Frame-Induced Packet Discarding [3], in which upon detection of loss of a threshold number of packets belonging to a video frame, the network attempts to discard all the remaining packets of that frame. In the problem of optimizing the quality of the transmitted video for a given cost function has been considered with leaky bucket constraints. Our work differs from theirs in that we are trying to optimize the QoS perceived by the user, rather than minimizing loss in general. In offline algorithms for optimal selective frame discard have been considered [4]. The notion of selective frame discard at the server has been introduced and the optimal selective frame discard problem using a QoS-based cost function has been defined.

II. Streaming Videos

Streaming video [5] is content sent in compressed form over the Internet and displayed by the viewer in real time. With streaming video or streaming media, a Web user does not have to wait to download a file to play it. Instead, the media is sent in a continuous stream of data and is played as it arrives. The user needs a *player*, which is a special program that uncompresses and sends video data to the display and audio data to speakers. A player can be either an integral part of a browser or downloaded from the software maker's Web site.

Major streaming video and streaming media technologies include RealSystem G2 from Real Network, Microsoft Windows Media Technologies (including its NetShow Services and Theater Server), and VDO. Microsoft's approach uses the standard MPEG compression algorithm for video. The other approaches use proprietary algorithms. (The program that does the compression and decompression is sometimes called the codec.) Microsoft's technology offers streaming audio at up to 96 Kbps and streaming video at up to 8 Mbps (for the NetShow Theater Server). However, for most Web users, the streaming video will be limited to the data rates of the connection (for example, up to 128 Kbps with an ISDN connection). Microsoft's streaming media files are in its Advanced Streaming Format (ASF).

Streaming video is usually sent from prerecorded video files, but can be distributed as part of a live broadcast "feed." In a live broadcast, the video signal is converted into a compressed digital signal and transmitted from a special Web server that is able to do multicast, sending the same file to multiple users at the same time.

This paper proposes and develops buffer management algorithms in specific environments, namely, those employing a FIFO scheduling regime. Our focus on FIFO scheduling follows from the following appealing features of such a queuing discipline: 1) it is simple, 2) it maintains the arrival order of incoming traffic, hence avoiding the need for mechanisms that deal with packet reordering, and 3) it provides simple and reliable delay bounds. These properties make FIFO especially attractive for delay and reordering-constrained streaming environments.

III.Design Procedure

In the proposed system, in order to overcome the issues of packet losses in the multimedia applications we provide two guidelines for designing buffer management algorithms, and demonstrate their effectiveness. Actually in the wireless medium, before transmission across the network the data frames are split into several smaller sized packets. In the streaming applications the packets are dependent to each other. The receiving side can make use of the data only if it receives all packets of a frame. The problem of ensuring that all packets of a frame arrive at the destination is crucial one. when considering real-time traffic, such as streaming multimedia traffic, where retransmission of missing packets is not feasible due to delay constraints posed by the application and also to increase the cost.

We suggest and extend buffer management algorithms in specific environments, namely, those employing a FIFO scheduling regime. Our focus on FIFO scheduling follows from the following appealing features of such a queuing discipline:

- It is simple
- It maintains the arrival order of incoming traffic, hence avoiding the need for mechanisms that deal with packet reordering
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The main reason for packet loss in networks is buffer overflows due to congestion. For data streams with packet dependencies, we must differentiate between the packet-level throughput, i.e., the amount of data delivered in terms of packets, and the effective goodput, i.e., the amount of data that can be decoded effectively at the receiving end. These two measures can be drastically different, e.g., the throughput may be high, while its goodput is very low.

Contribution

The main contributions are as follows:

We provide two design guidelines for algorithms, namely,

- No-regret: Once a frame has a packet admitted to the buffer, one should make every attempt possible to deliver the complete frame
- Ensure-progress: Strive to deliver a complete frame as soon as possible.
- We devise a buffer management algorithm, WEIGHTPRIORITY that follows these guidelines. We analyze the performance of our algorithm, and show that for any arrival traffic the ratio between its performance and that of an optimal algorithm is always bounded.

Algorithm WEIGHTPRIORITY

1: If f_i^m i is alive and $L_t\ (f_i^m)$ yields to $A_t\ (f_i^m)$ then

2: let $D_t(f_i^m)\ \epsilon$ p($L_t\ (f_i^m)$) be the minimal size set that yields to $A_t\ (f_i^m)$

3: Preempt $D_t(f_i^m)$

4: accept $A_t\ (f_i^m)$

5: else

6: reject $A_t\ (f_i^m)$

7: drop $R_t(f_i^m) \cap f_i^m$

8: endif

IV. CONCLUSION AND FUTURE WORK

The buffer management scheme is used for avoiding the packet losses. Because in the streaming applications each and every packets depends on other packets. So, if there is a loss in any packet the whole sequence is useless. To avoid these problems we use buffer management schemes to discard some of the packets that depends on the ranking schemes. We provided guidelines for the design of such algorithms, and analysed the performance of one such algorithm, both from a worst case competitive approach, as well as by a simulation study. We provided guarantees on its performance under any traffic conditions by proving it has a bounded competitive ratio. We also showed that the competitive ratio of any algorithm for our problem might degrade linearly as a function of the number of streams in the traffic.

Securing multimedia data has become of utmost importance especially in the applications related to military purposes. Using innovative encryption algorithms for video sequences are necessary for protect the data. This can be done in future work.

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